REMARKS/ARGUMENTS

This case has been carefully reviewed and analyzed in view of the Office Action dated 12 September 2008. Responsive to the rejections made in the Office Action, Independent Claims 1-2 have been amended. Thus, Claims 1-16 remain pending.

In the Office Action, the Examiner objected to the Drawings under 37 C.F.R. § 1.83(a) as not showing every feature of the invention specified in the Claims. Specifically, the Examiner pointed out that the Drawings fail to show the "local SIP agent client" which is recited in Claims 1 and 2.

Responsive to the objection to the Drawings, the Applicant has amended the terminology used in Claims 1 and 2, e.g. the "local SIP agent client" in both Claims 1 and 2 and replaced it with "SIP agent client of a local device". This claim recitation is completely supported by Fig. 3 of the Application as well as the pertaining disclosure on page 8, line 13, - page 9, line 16, of the Specification.

The Specification has been amended on p. 6 thereof to further promote the compliance between the Specification, Drawings and the Claims.

Additionally, the Examiner objected to the Drawings since, as the Examiner asserted, the Specification failed to disclose "a local SIP agent client … to convert an analog voice signal of the local device into a digital signal … and to convert a digital signal sent from the remote SIP agent client into analog voice signal …" as recited in Claims 1 and 2.

The attention of the Examiner is respectfully drawn to p. 9, line 3 from the bottom – p. 10, line 2 of the original Specification which clearly communicates that "the SIP agent client 52 … is directly connected with an IP phone for compressing and converting the voice signal of the local device 2 into digital signal or decompressing and converting the digital signal into voice signal …". The Applicant thus believes that there is no discrepancy between the Specification and Fig. 3 of the Drawings and the subject matter claimed.

It is believed that, as a result of amendments to the Claims and the Specification, the objection to the Drawings under 37 C.F.R. § 1.83(a) has been obviated.

Further, in the Official Action, the Examiner objected to Claims 1 and 2 because of informalities found therein. The Applicant has amended the Claims as was suggested by the Examiner and these objections are believed to be obviated by this Amendment.

Additionally, Claims 1-16 were rejected under 35 U.S.C. § 112, first paragraph, because the Examiner did not find that the Specification reasonably provides an enablement for a local SIP agent client to convert an analog voice signal into a digital signal, and to convert a digital signal sent from the remote SIP agent client into an analog voice signal.

The Applicant respectfully disagrees with this assertion of the Examiner, and respectfully submits that this feature of the invention which is covered in

Claim 1 is fully supported by the Specification on page 9, line 3 from the bottom – page 10, line 2 and related Fig. 3 of the subject Patent Application.

Claim 2 covers an alternative embodiment of the system of the present invention, shown in Fig. 2, where the voice process module converts the analog voice signal generated by the IP phone into digital signal and also converts the digital signal sent from the SIP processing module into the analog voice signal.

Further, the Examiner pointed out that the Specification does not enable "a local SIP agent client for executing at least one SIP agent client program ..." and "... an SIP call server for executing at least one SIP call server program ..." as recited in Claims 1 and 2.

It is respectfully submitted that the features of execution of the related programs in the SIP agent client and in the SIP call server are inherent features for the SIP agent client and SIP call server. However, in order to facilitate the prosecution of the Patent Application in question, the Applicant has amended Claims 1 and 2 to remove the features in question, e.g. - - executing at least one SIP agent client program and executing at least one SIP call server program - - from Claims 1 and 2.

In addition, the Examiner pointed out that the recitation "being SIP structure" for the SIP call server in the Claims is recited only in the preamble and therefore this recitation has not been given patentable weight. Accordingly, Claims 1 and 2 have been amended to recite "SIP call server being based on an

SIP structure" in the body of the Claims. Therefore, this recitation now may be given a patentable weight; and the same is respectfully requested.

It is respectfully submitted that in view of the above discussion and corrections to the Claims 1 and 2, that the rejection under 35 U.S.C. § 112, first paragraph, is believed to be obviated.

Additionally, Claims 1-16 were rejected under 35 U.S.C. § 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which Applicant regards as the invention. Specifically, with regard to Claims 1 and 2, the Examiner was unclear what is "the network apparatus" and what is the association of this terminology with the claimed network system.

The Applicant has amended Claims 1 and 2 to replace the terminology "the network apparatus" with the language "the network system". It is believed that by this amendment, the rejection under 35 U.S.C. § 112, second paragraph, has been removed.

Further in the Official Action, the Examiner rejected Claims 1 and 2 under 35 U.S.C. § 103(a) as being unpatentable over Osterhout, et al., U.S. Patent 6,965,614, in view of Lazarus, et al., U.S. Patent Application Publication 2004/0234059, and Donovan, et al., U.S. Patent 6,512,812.

The Examiner did not give reason for rejection of Claims 3-16 based on prior art. The Applicant respectfully requests whether the Examiner found allowable subject matter in Claims 3-16.

Prior to discussion of the prior art relied upon by the Examiner, and the distinguishing features of the present invention over the cited references, it is believed to be beneficial to briefly review the structure of the present invention covered by the Independent Claims 1 and 2 and other pending Claims of the subject Patent Application.

The present invention is directed to an SIP telecommunication network which is modified into a "private" SIP telecommunication network by inclusion of a network system 1 between a user (local device) and SIP agent client 53 of remote device 4. The network system 1 is integrated with SIP call server 51 and SIP agent client 52 of the local device, and operates in accordance with the Session Initiation Protocol (SIP). The network system 1 is coupled between the local device 2 and a network 32 for connecting with at least one remote SIP agent client 53.

The network system 1 includes at least one local connecting port 11 for coupling with the local device 2, and a remote connecting port 12 for coupling with the network. The network system 1 further includes an SIP processing module 14 electrically connected with the local connecting port and remote

connecting port 12, as shown in Fig. 3 and disclosed on pp. 9-10 of the Specification.

The SIP processing module 14 may be directly connected with the local device, such as for example, an IP phone, so that the local SIP agent client 52 of the SIP processing module executes the SIP agent client program to convert an analog voice signal of the local device into a digital signal and to send the digital signal to the remote SIP agent client 53, or to convert the digital signal received from the remote SIP agent client 53 into an analog voice signal and then send the voice signal to the local device.

The SIP processing module 14 also includes an SIP call server 51 for executing at least one SIP call server program to facilitate the local SIP agent client and the remote SIP agent client's bidirectional telecommunication each with the other by voice upon the local SIP agent client and the remote SIP agent client perform SIP registry and the locations of the local SIP agent client and the remote SIP agent client are linked.

In an alternative implementation, shown in Fig. 2, the telecommunication system includes the network system 1 which has an IP phone connecting port for coupling with the IP phone, and a voice processing module coupled between the IP phone connecting port and the SIP processing module to perform A/D or D/A conversion of the voice signal.

Osterhout, et al., the main reference cited by the Examiner, is directed to a communication system between different types of devices. In the cited system, an SIP gateway 32 provides ports to various peripheral devices. A network element is coupled to the data network 12 to establish Session Initiation Protocol (SIP) sessions with the gateway 32. Once an SIP session is established, communications may occur between the network element and the peripheral devices. SIP messaging is exchanged between the network cloud 14 and the gateway 32. One type of peripheral device includes a USB device. The gateway 32 provides conversion between the SIP messaging and the USP commands and data.

It is respectfully submitted that, in contrast to the claimed device, in Osterhout, et al., the SIP gateway 32 is not integrated with an SIP agent client of a peripheral device nor with the SIP call server. While in the telecommunication network of the present invention, the network system 1 is "... integrated with an SIP call server and an SIP agent client of a local device ...".

Further, Osterhout, et al. fails to teach an SIP agent client of a peripheral device "converting an analog voice signal of the local device into a digital signal and sending the digital signal to the SIP agent client of ... remote device, and converting a digital signal sent from the SIP agent client of said ... remote device into analog voice signal ... to the local device" as it is recited in Claim 1 among its other limitations.

In further contradistinction with the present invention, Osterhout, et al.'s system fails to teach "a voice processing module electrically connected with … IP phone connecting port and … SIP processing module" and converting an analog voice signal of the IP phone into a digital signal, or converting the digital signal of the SIP processing module into an analog voice signal" as it is claimed (inter alia) in Claim 2.

Lazarus, et al., another reference cited by the Examiner, is directed to telephone conference bridging within a residential gateway. The Examiner cited this reference for teaching a cable modem in communications gateway used to convert the received RF signal into digital based band signals and digital based band signals into RF signals for transmission [0007].

It is respectfully submitted, that, in contrast to the present invention, Lazarus, et al. <u>is not</u> an SIP structure and is <u>not based</u> on the SIP communication protocol. Therefore, Lazarus, et al. is not intended and does not base his system on an SIP processing module which has an SIP agent client converting an analog voice signal into the digital signal, or converting a digital signal received from the remote SIP agent client into analog voice signal, as takes place in the present invention.

Nor does the Lazarus, et al. reference teach a voice processing module electrically connected with the IP phone connecting port and the SIP processing module and operating to perform A/D or D/A voice signal conversions.

Additionally, Lazarus, et al. never contemplates the integration of the agent client and the call server with the local device originating the call, as is taught in the present Application.

Donovan, et al., another secondary reference cited by the Examiner, is directed to a system for releasing a voice response unit from a protocol session. In Donovan, et al., a call originator, acting as a user agent client in accordance with the Session Initiation Protocol (SIP), issues messages to establish a first call-leg with the Voice Response Unit (VRU). The VRU performs digit collection to obtain information to authenticate the call originator and to authorize the voice call. Based upon the issued messages from the call originator, the VRU establishes a second call-leg with a call terminator. The VRU is released from the voice call after binding the call-legs to connect the call originator to the call terminator.

It is respectfully submitted that, in contrast to the subject telecommunication network, in Donovan, et al. the user agent client or the SIP call server are never integrated with the system of the call originator and therefore Donovan, et al. is not capable of a "private" SIP telecommunication between the user originators and the user terminators, as is provided in the present invention. Due to the lack of the integration of the user agent client and the server of the caller originator, the Voice Response Unit must obtain information to authenticate the call originator, to authorize the voice call, and to charge the call originator accordingly. This scheme is similar to the one used in the Applicant's admitted

prior art, as described in pertinent sections of the original Specification, and which is obviated in the present invention by inclusion of a network system between a user and the SIP agent client of remote devices and which is integral with the SIP call server and SIP agent client of the local device. This feature is completely missing in Donovan, et al.

Further, in contrast to the present telecommunication system, Donovan, et al. does not disclose the SIP agent client of the local device capable of converting an analog voice signal of the local device into a digital signal to be sent to the remote SIP agent client, or to convert a digital signal sent from the remote SIP client into an analog voice signal and sending the analog signal to the local device, as it is covered in Claim 1. Nor does Donovan, et al. teaches a voice processing module electrically connected to the phone connecting port and the SIP processing module to convert an analog voice signal of the IP phone into a digital signal or to convert the digital signal of the SIP processing module into an analog voice signal, as it is claimed in Claim 2.

Summarizing the arguments presented in previous paragraphs, it is believed that none of the references cited by the Examiner, taken singly or in combination, teaches a telecommunication network which includes (inter alia):

"... network system integrated with an SIP call server and an SIP agent client of a local device ...", or

"... SIP agent client of the local device converting an analog voice signal of the local device into a digital signal and sending the digital signal to the SIP agent client of ... remote device, and converting a digital signal sent from the SIP agent client of ... remote device into analog voice signal and sending the analog voice signal to the local device ...", as claimed in Claim 1.

Further, none of the references cited by the Examiner, taken solely or in combination, teaches (among others) the following elements:

"... a network system integrated with an SIP call server and an SIP agent client of a local device ...", or

"... a voice processing module electrically connected with ... IP phone connecting port and ... SIP processing module, the voice processing module converting an analog voice signal of the IP phone into a digital signal, or converting the digital signal of the SIP processing module into an analog voice signal ...", as claimed in Claim 2.

The claimed features are not disclosed, suggested or rendered obvious by any of the references cited by the Examiner, nor by their combination.

As the combination of Osterhout, et al., Lazarus, et al. and Donovan, et al. fail to disclose or suggest the concatenation of limitations that define the invention of the subject Patent Application, as now claimed in Claims 1 and 2, they are not believed to make obvious that invention. Accordingly, the allowance of newly-amended Independent Claims 1 and 2 is respectfully requested.

It is believed that Claims 3-16, dependent on Claim 1, add further patentably distinct limitations, and are patentably distinct for at least the same reasons as presented in the discussion of Independent Claim 1, and therefore are believed to be allowable.

For all of the foregoing reasons, it is now believed that the subject Patent Application has been placed in condition for allowance, and such action is respectfully requested.

If there are any further charges associated with this filing, the Director of Patents and Trademarks is hereby authorized to charge Deposit Account #18-2011 for such charges.

Respectfully submitted,
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Dated: 1/7/09

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